I claim:

1. A method for time scaling and/or pitch shifting an audio signal, comprising

analyzing said audio signal using multiple psychoacoustic criteria to identify a region of the audio signal in which the omission of a portion of the audio signal or the repetition of a portion of the audio signal is inaudible or minimally audible,

selecting a splice point in said region of the audio signal,

deleting a portion of the audio signal beginning at the splice point or repeating a portion of the audio signal ending at the splice point, and

reading out the resulting audio signal at a rate that yields a desired audio signal time duration and a desired time scaling and/or pitch shifting.

2. A method for time scaling and/or pitch shifting an audio signal represented by samples, comprising

analyzing said audio signal using multiple psychoacoustic criteria to identify a region of the audio signal in which the omission of a portion of the audio signal or the repetition of a portion of the audio signal is inaudible or minimally audible,

selecting a splice point in said region of the audio signal, thereby defining a leading segment of the audio signal that leads the splice point,

selecting an end point spaced from said splice point, thereby defining a trailing segment of the audio signal that trails the endpoint, and a target segment of the audio signal between the splice and end points,

joining said leading and trailing segments at said splice point, thereby decreasing the number of audio signal samples by omitting the target segment when the end point has a higher sample number than said splice point, or increasing the number of samples by repeating the target segment when the end point has a lower sample number than said splice point, and

reading out the joined leading and trailing segments at a rate that yields a desired audio signal time duration and a desired time scaling and/or pitch shifting.

3. The method of claim 2 wherein:

a time duration the same as the original time duration results in pitch shifting the audio signal,

a time duration decreased by the same proportion as the relative change in the reduction in the number of samples, in the case of omitting the target segment, results in time compressing the audio signal,

a time duration increased by the same proportion as the relative change in the increase in the number of samples, in the case of repeating the target segment, results in time expanding the audio signal,

a time duration decreased by a proportion different from the relative change in the reduction in the number of samples results in time compressing and pitch shifting the audio signal, and

a time duration increased by a proportion different from the relative change in the increase in the number of samples results in time expansion and pitch shifting the audio signal.

4. The method of claim 2 wherein the end point is also selected to be in said region.

5. The method of claim 2 wherein analyzing said audio signal using multiple psychoacoustic criteria includes analyzing said audio signal to identify a region of the audio signal in which the audio satisfies at least one criterion of a group of psychoacoustic criteria.

6. The method of claim 5 wherein said psychoacoustic criteria include at least one of the following:

the identified region of said audio signal is substantially premasked or postmasked as the result of a transient,

the identified region of said audio signal is substantially inaudible,

the identified region of said audio signal is predominantly at high frequencies, and

the identified region of said audio signal is a quieter portion of a segment of the audio signal in which a portion or portions of the segment preceding and/or following the region is louder.

7. The method of claim 6 wherein the criterion that the region of said audio is predominantly at high frequencies is based on frequencies above about 10 to 12 kHz.

8. The method of claim 5 wherein said group of psychoacoustic criteria are arranged in order of the increasing audibility of artifacts resulting from the joining of the leading and trailing segments at said splice point.

9. The method of claim 8 wherein said region is identified when the highest ranking psychoacoustic criterion, the criterion leading to the least audible artifacts in said group is satisfied.

10. The method of claim 8 wherein the top-ranked psychoacoustic criterion is that the region of said audio signal is substantially premasked or masked as the result of a transient.

11. The method of claim 8 wherein said psychoacoustic criteria include at least the following four criteria, ranked in the following order:

the identified region of said audio signal is substantially premasked or postmasked as the result of a transient,

the identified region of said audio signal is substantially inaudible,

the identified region of said audio signal is predominantly at high frequencies, and

the identified region of said audio signal is a quieter portion of a segment of the audio signal in which a portion or portions of the segment preceding and/or following the region is louder.

12. The method of claim 11 wherein the criterion that the region of said audio is predominantly at high frequencies is based on frequencies above about 10 to 12 kHz.

13. The method of claim 5 wherein the end point is also selected to be in said region.

14. The method of any one of claims 2 wherein said step of joining said leading and trailing segments at the splice point includes crossfading the leading and trailing segments.

15. The method of claim 14 wherein the crossfading is a linear crossfading.

16. The method of claim 14 wherein the crossfading is a nonlinear crossfading.

17. The method of claim 16 wherein the nonlinear crossfading is in accordance with a Hanning window.

18. The method of claim 16 wherein the nonlinear crossfading is in accordance with a Kaiser-Bessel window.

19. The method of claim 14 wherein the length of the crossfade resulting from crossfading the leading and trailing segments is variable.

20. The method of claim 19 wherein the length of the crossfade resulting from crossfading the leading and trailing segments is selected to minimize audible splicing artifacts.

21. The method of claim 19 wherein the crossfading is a linear crossfading.

22. The method of claim 19 wherein the crossfading is a nonlinear crossfading.

23. The method of claim 22 wherein the nonlinear crossfading is in accordance with a Hanning window.

24. The method of claim 22 wherein the nonlinear crossfading is in accordance with a Kaiser-Bessel window.

25. The method of claim 2, wherein in the case of decreasing the number of audio signal samples by omitting the target segment, said end point is selected by autocorrelating a segment of audio trailing the splice point.

26. The method of claim 25 wherein said end point is selected by autocorrelating a segment of audio trailing the sample point up to a maximum processing point and selecting an end point substantially at the point of maximum autocorrelation that is greater than a minimum processing point.

27. The method of claim 26 wherein said maximum processing point is fixed relative to said splice point.

28. The method of claim 26 wherein said minimum processing point is variable relative to said splice point.

29. The method of claim 28 wherein said minimum processing point is variable relative to said splice point in response to signal characteristics near said splice point.

30. The method of claim 25 wherein said autocorrelation is based on the time-domain characteristics of the audio signal.

31. The method of claim 30 wherein the time domain characteristics of the audio signal are weighted according to human hearing sensitivity.

32. The method of claim 25 wherein said autocorrelation is based on the phase characteristics of the audio signal.

33. The method of claim 32 wherein said autocorrelation is based on the instantaneous phase characteristics of the audio signal.

34. The method of claim 25 wherein said autocorrelation is based on the phase and time-domain characteristics of the audio signal.

35. The method of claim 34 wherein the time domain characteristics of the audio signal are weighted according to human hearing sensitivity.

36. The method of claim 34 wherein said autocorrelation is based on the instantaneous phase and time-domain characteristics of the audio signal.

37. The method of claim 36 wherein the time domain characteristics of the audio signal are weighted according to human hearing sensitivity.

38. The method of claim 2, wherein in the case of increasing the number of audio signal samples by repeating the target segment, said end point is selected by cross correlating segments of audio leading and trailing the splice point.

39. The method of claim 38 wherein said end point is selected by cross correlating a segment of audio leading the splice point up to a first maximum processing point and a segment of audio trailing the splice point up to a second maximum processing point and selecting an end point in the segment leading the splice point substantially at the point of maximum cross correlation that is greater than a minimum processing point.

40. The method of claim 39 wherein said first and second maximum processing points are fixed relative to said splice point.

41. The method of claim 40 wherein said first and second maximum processing points are the same relative to said splice point.

42. The method of claim 39 wherein said minimum processing point is variable relative to said splice point.

43. The method of claim 42 wherein said minimum processing point is variable relative to said splice point in response to signal characteristics near said splice point.

44. The method of claim 38 wherein said cross correlation is based on the time-domain characteristics of the audio signal.

45. The method of claim 44, wherein the time domain characteristics of the audio signal are weighted according to human hearing sensitivity.

46. The method of claim 38 wherein said autocorrelation is based on the phase characteristics of the audio signal.

47. The method of claim 46 wherein said autocorrelation is based on the instantaneous phase characteristics of the audio signal.

48. The method of claim 38 wherein said autocorrelation is based on the phase and time-domain characteristics of the audio signal.

49. The method of claim 48, wherein the time domain characteristics of the audio signal are weighted according to human hearing sensitivity.

50. The method of claim 48 wherein said autocorrelation is based on the instantaneous phase and time-domain characteristics of the audio signal.

51. The method of claim 50, wherein the time domain characteristics of the audio signal are weighted according to human hearing sensitivity.

52. A method for time scaling and/or pitch shifting multiple channels of audio signals, comprising

analyzing each of said audio signals using at least one psychoacoustic criterion to identify at least one region in each of the audio signals in which the omission of a portion of the audio signal or the repetition of a portion of the audio signal is inaudible or minimally audible,

selecting a common splice point in one of said regions in each of the audio signals, wherein the splice points in the multiple channels of audio signals are selected to be substantially aligned with one another,

deleting a portion of each audio signal beginning at the common splice point or repeating a portion of the audio signal ending at the common splice point, and

reading out the resulting audio signals at a rate that yields a desired time duration for the multiple channels of audio and a desired time scaling and/or pitch shifting for the multiple channels of audio.

53. A method for time scaling and/or pitch shifting multiple channels of audio signals, each signal represented by samples, comprising

analyzing each of said audio signals using at least one psychoacoustic criterion to identify at least one region in each of the audio signals in which the omission of a portion of the audio signal or the repetition of a portion of the audio signal is inaudible or minimally audible,

selecting a common splice point in one of said regions in each of the audio signals, thereby defining a leading segment of the audio signal that leads the splice point, wherein the splice points in the multiple channels of audio signals are selected to be substantially aligned with one another,

selecting an end point spaced from said splice point in each of the audio signals, thereby defining a trailing segment of the audio signal trailing the endpoint and a target segment of the audio signal between the splice and end points, wherein the end points in

the multiple channels of audio signals are selected to be substantially aligned with one another,

joining said leading and trailing segments at said splice point in each of the audio signals, thereby decreasing the number of audio signal samples by omitting the target segment when the end point has a higher sample number than said splice point, or increasing the number of samples by repeating the target segment when the end point has a lower sample number than said splice point, and

reading out the joined leading and trailing segments in each of the audio signals at a rate that yields a desired time duration for the multiple channels of audio and a desired time scaling and/or pitch shifting for the multiple channels of audio.

54. The method of claim 53, wherein:

a time duration the same as the original time duration results in pitch shifting the audio signals,

a time duration decreased by the same proportion as the relative change in the reduction in the number of samples, in the case of omitting the target segment, results in time compressing the audio signals,

a time duration increased by the same proportion as the relative change in the increase in the number of samples, in the case of repeating the target segment, results in time expanding the audio signals,

a time duration decreased by a proportion different from the relative change in the reduction in the number of samples results in time compressing and pitch shifting the audio signals, and

a time duration increased by a proportion different from the relative change in the increase in the number of samples results in time expansion and pitch shifting the audio signals.

55. The method of claim 53 wherein said selecting a common splice point selects a splice point in each of said audio signals, one or more of which splice points may not not be coincident with one or more other splice points, and selects one of said splice points as the common splice point.

56. The method of claim 55 wherein said selecting a common splice point selects a common splice point from among said splice points using the psychoacoustic criteria employed to identify said regions.

57. The method of claim 56 wherein said selecting a common splice point selects a common splice point from among said splice points by also taking into account cross-channel effects using at least one psychoacoustic criterion.

58. The method of claim 53 wherein said selecting a common splice point identifies portions of said audio signals in which identified regions overlap, and selects a common splice point in the overlapping portion of said identified regions.

59. The method of claim 58 wherein said selecting a common splice point selects a common splice point in the overlapping portion of said identified regions using the psychoacoustic criteria employed to identify said regions.

60. The method of claim 58 wherein said selecting a common splice point selects a common splice point in the overlapping portion of said identified regions by also taking into account cross-channel effects using at least one psychoacoustic criterion .

61. The method of claim 3 wherein said selecting a common splice point selects a common splice point using the psychoacoustic criteria employed to identify said regions.

62. The method of claim 61 wherein said selecting a common splice point selects a common splice point by also taking into account cross-channel effects using at least one psychoacoustic criterion .

63. The method of claim 53 wherein the end point is also selected to be in said region in each of the audio signals.

64. The method of claim 53 wherein analyzing said audio signal using at least one psychoacoustic criterion to identify at least one region in each of the audio signals in which the omission of a portion of the audio signal or the repetition of a portion of the audio signal is inaudible or minimally audible includes analyzing said audio signal to identify a region of the audio signal in which the audio satisfies at least one criterion of a group of psychoacoustic criteria.

65. The method of claim 64 wherein said psychoacoustic criteria include at least one of the following:

the identified region of said audio signal is substantially premasked or postmasked as the result of a transient,

the identified region of said audio signal is substantially inaudible,

the identified region of said audio signal is predominantly at high frequencies, and

the identified region of said audio signal is a quieter portion of a segment of the audio signal in which a portion or portions of the segment preceding and/or following the region is louder.

66. The method of claim 65 wherein the criterion that the region of said audio is predominantly at high frequencies is based on frequencies above about 10 to 12 kHz.

67. The method of claim 64 wherein said group of psychoacoustic criteria are arranged in order of the increasing audibility of artifacts resulting from the joining of the leading and trailing segments at said splice point.

68. The method of claim 67 wherein said region is identified when the highest-ranking psychoacoustic, the criterion leading to the least audible artifacts in said group is satisfied.

69. The method of claim 67 wherein the top-ranked psychoacoustic criterion is that the region of said audio signal is substantially premasked or masked as the result of a transient.

70. The method of claim 67 wherein said psychoacoustic criteria include at least the following four criteria, ranked in the following order:

the identified region of said audio signal is substantially premasked or postmasked as the result of a transient,

the identified region of said audio signal is substantially inaudible,

the identified region of said audio signal is predominantly at high frequencies, and

the identified region of said audio signal is a quieter portion of a segment of the audio signal in which a portion or portions of the segment preceding and/or following the region is louder.

71. The method of claim 70 wherein the criterion that the region of said audio is predominantly at high frequencies is based on frequencies above about 10 to 12 kHz.

72. The method of claim 64 wherein the end point is also selected to be in said region.